Speaker Building Theory and Implementation

A Music Technology Thesis (MMP495)

By
Andrew Ayers

Thesis Advisor
Dr. Michael Pounds

Michael L. Pounds

Ball State University
Muncie, Indiana

December 2012

Expected Date of Graduation
December 2012

Abstract

Speaker design and construction is an interesting field that combines knowledge of acoustics, electronics, and carpentry with the ever-subjective target of making things sound good. Despite the wide range of speakers available from numerous companies, with some care and consideration the hobbyist builder can still create products of similar or better quality for the a similar price. I will start with a discussion of the main components of the modern near-field loudspeaker, giving a background in the scientific principles at work, and move to the most commonly seen configurations of those components. After discussing the background and theory, I will then move on to a step-by-step walkthrough of the process that I used to turn those theories into a usable product.

Acknowledgements

I would like to that Dr. Pounds for the guidance and helpful suggestions that he has provided throughout this project.

I would like to thank the staff at the College of Architecture and Planning 'FAB-LAB' for their immeasurable help with the fabrication part of this project.

I would like to thank the community of the Parts Express web forum, especially Jeff Bagby, and Paul Carmody, for their kind help and prompt responses when I had questions.

Table of Contents

Introduction

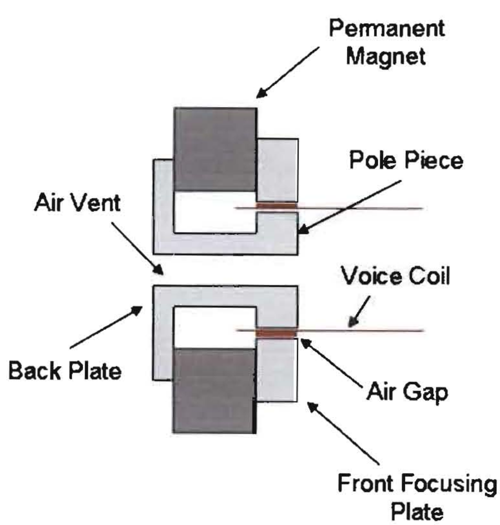
The idea of designing and building loudspeakers has a certain mystique that interests many people of different academic backgrounds. Of all the common household electronics, loudspeakers are just complex enough to present a challenge, but simple enough to entice do-it-yourself-ers. Given my academic background in music technology and physics, it seemed an appropriate capstone project to break through the mystique and find the science of these devices that we use everyday. My findings showed that the overall operation of a loudspeaker is best described as the interaction of three main parts, the drivers, the cabinet, and the crossover. In the course of this paper I will discuss the theory and operation of these elements as well as how that knowledge influenced the decisions I made about my design. I will also talk about the process I used to create and build my design. The final section of my thesis will consist of an objective and subjective evaluation of my final product.

The Drivers

Although there are many different variations on the standard electrodynamic speaker driver, the most commonly used transducer is of the moving coil-permanent magnet type. A transducer is any device that converts one type of energy to another, so in this case a loudspeaker is a device that converts electrical energy into acoustic energy. The standard speaker can be broken up into three systems that govern its functions:

1. The Motor System – this is made up of the magnet, pole piece, frontplate/gap, and voice coil.
2. The Diaphragm – this is usually a cone and a dust cap, or sometimes a one-piece cone.
3. The Suspension System – this is made up of the spider and the surround.

The motor system is made up of five essential parts. These are the frontplate, pole piece, backplate, magnet, and voice coil. The backplate, frontplate, and pole piece are all made up of a magnetically permeable material, such as iron, so that the magnetic fields, which allow the device to function, will not be hindered. When an AC signal is applied to the voice coil, it creates an intense magnetic field



that reacts with the field of the permanent magnet, setting the voice coil in motion. The voice coil is attached to the diaphragm system, causing it to move as well.[1] There are several variables used to define the performance of a motor system that are usually provided by the manufacturer so that the consumer may evaluate the capabilities of the driver. These five parameters are: magnet weight, voice-coil length, Bl, voice-coil diameter, and $X_{max.}$ The magnet weight is quite simply the weight of the magnet, which can be quite heavy. The voice-coil length gives the length of the wire wrapped around the voice-coil former.[2] The next variable Bl refers to the strength of the motor in Tesla*Meters/Newton. This is derived by taking the number of turns of the voice coil (L) and multiplying it by the magnetic flux density (B) in the gap between the frontpiece and the pole piece.[3] The last variable mentioned above is $X_{max.}$ This is a measurement in millimeters that refers to how far the voice coil can safely move in and out of the gap. When manufacturers add this parameter, they are usually referring to how far the voice coil can move before it loses an acceptable level of linearity. As the voice coil moves out of the gap, less turns of wire in the gap means less force exerted by the motor, which causes the loss of linearity. Sometimes manufacturers will also add an additional variable called $X_{mech,}$ which denotes the actual physical distance that the voice-coil could move without hitting the backplate.

[1] Dickason 3-4
[2] Alden 11
[3] Dickason 3
[4] Dickason 7
[5] Dickson 11

The diaphragm is made up of the cone and a dust cap. The cone is attached to the voice coil, and is responsible for turning the movement of the voice coil into the movement of air. The dust cap only serves to keep any stray particulates

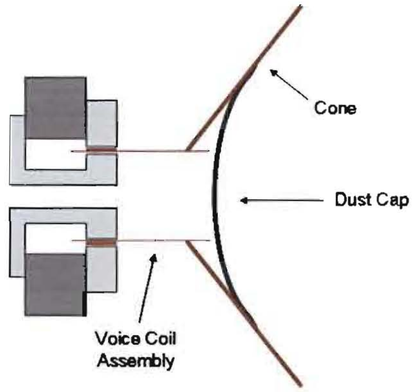


Figure 2 - Diaphragm

from getting down into the motor. The ideal cone would be both infinitely rigid and have no mass, but in reality they will have mass and flex to some degree based upon the material that they are made from. The flexing of the cone will have an impact on the tonal quality or sound of the driver.[4] As technology continues to improve manufacturers have found new materials to improve the ratio of lightness to stiffness. While speaker cones were originally made from paper, they are now made from materials like polypropylene, woven fiberglass, carbon fiber, metal alloys, or Kevlar. Each material has a set of advantages and disadvantages that make it uniquely suited for certain applications. For example, metal alloy drivers such as the aluminum woofers that I selected for my design have exceptional response over a certain range, but begin to exhibit unwanted breakup modes that must be compensated for with the crossover outside that range. Paper or treated paper tends to have a forgiving frequency response, but at the expense of the rigidity and clear transients of harder materials.[5] The only measurements that a manufacturer provides about the diaphragm are the diameter of the cone and the effective

[4] Dickason 7
[5] Dickson 11

surface area (S_d) of the cone. Both of these don't mean much on their own, but will be used in calculations with the surround.[6]

The last system that makes up a driver is the suspension. The suspension consists of the surround and the spider. These pieces serve two main purposes. They serve to keep the cone centered, and to provide the restorative forces necessary to return the cone back to its original position after excursion. The surround also

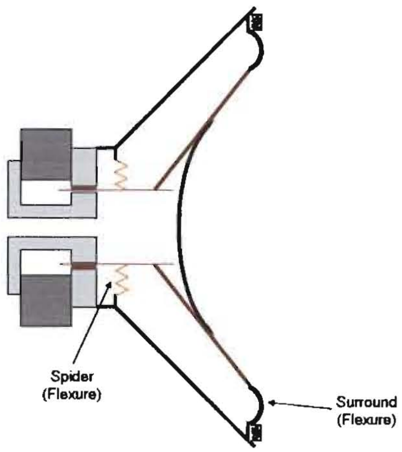


Figure 3 - Suspension System

serves to keep the speaker sealed, dampen any unwanted modes of vibration, and prevent reflections back down the cone. The surround provides less restorative force than the spider, usually at a ratio of 20% to 80%. The surround is usually made of rubber or foam, with rubber having superior damping qualities but is more expensive to manufacture.[7] There are several important parameters provided by manufacturers that relate to the suspension system. M_{MS} is the mass of the driver's moving mechanical system including the cone, surround, and dust cap. C_{MS} is a measure of the mechanical compliance of the suspension system. In this case, the word compliance can be understood as the inverse of stiffness. The C_{MS} value is important for calculating a volume of air that has the same compliance as the suspension system. The V_{AS} value becomes

[6] Alden 10
[7] Dickason 11

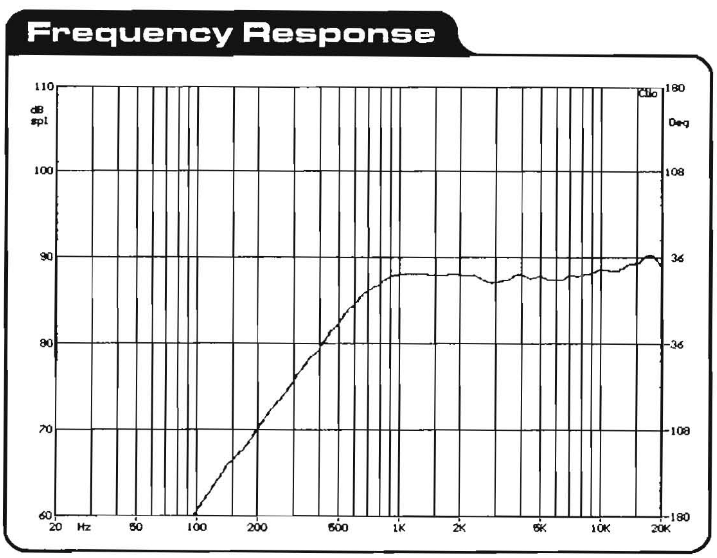
important in designing a sealed box, which we will come to later on. The formula for calculating V_{AS} from C_{MS} is:

$$V_{AS} = \rho * c^2 * C_{MS} * {S_D}^2$$

Equation 1 - Calculating Vas

In the above equation ρ and c are constants for the density of air and the speed of sound in air respectively. R_{MS} is a value representing the mechanical resistance summed through all of the suspension losses. This value is different from an electrical resistance and is given in kg/s instead of ohms. The last important value for the suspension system is Q_{MS}. This is a measure of the mechanical damping that the cone exhibits. Although there are other important parameters of the driver's operation that will come up in other sections of this paper, these will be addressed as encountered instead of at this point.

When I was choosing the drivers for my design, I knew that I wanted to build a two-way system that had a fairly flat response across the listening range. A two-way system means that the loudspeaker uses two drivers in combination. Three-way systems are just as common, but the electronics and the price point were more daunting for my first try at building my own speakers. Ideally I was aiming to produce something in the like of any of the many kinds of



two-way reference monitors that are commonly seen throughout studios.

The beginning of my selection process began with me seeing the Dayton Audio RS28F-4 on www.partsexpress.com that was on clearance from $95 down to $45. After investigating, I found that the RS28F-4 had a smooth frequency response with exceptional quality above 10kHz and extending down to a resonant point around 500Hz as is shown by the figure. As shown by the model number this is the four ohm version of this driver, which should pose no problem for any modern stereo receiver, although some old tube amplifiers are very sensitive to incorrect impedances.

Having made a decision on the tweeter, I now needed to find a woofer that worked well with it. A good rule of thumb is to have an octave to an octave and a half of useable overlap between the two drivers.[8] The website that I was shopping through suggested to me that many other people had also looked at the Dayton Audio RS180 woofer. Looking at the woofer's response, we can see that the region

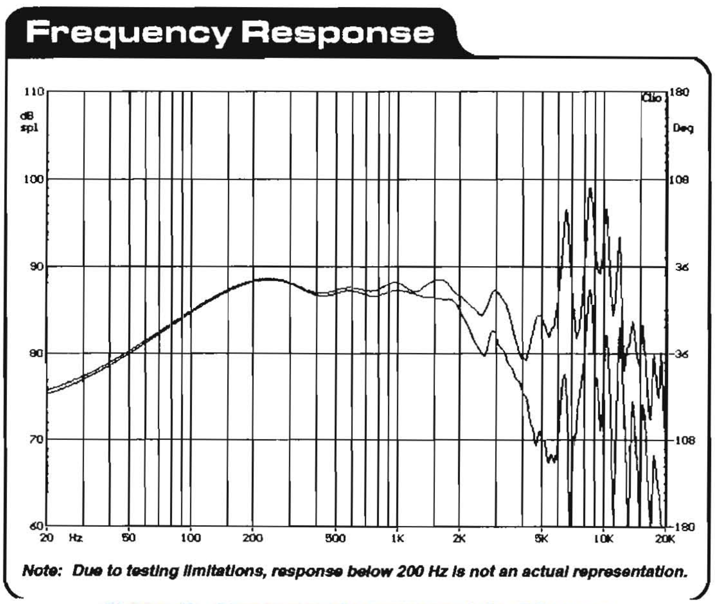


Figure 5 - Frequency Response of the RS180-4

[8] Alden Somewhere?

between 500Hz and 2000Hz is fairly flat in both drivers. The RS180-4 is an aluminum driver though, and this does manifest itself in the breakup modes above 5000Hz shown in the graph. After I had already bought both drivers, I received some feedback from the Parts Express user forum that there is also a paper cone version of this driver that is much easier to work with. Having selected my drivers, I moved on to planning the box. The process doesn't have to start with the drivers, in fact many people find enclosures that they want to reuse and then find drivers to fit the box. Starting with the drivers is just the way that I planned my design.

The Enclosure

The next part of the loudspeaker that will be discussed is the enclosure. While there are many different kinds of enclosures, I will only focus on sealed and ported boxes. Other kinds of enclosures worth mentioning are the bandpass box, the horn-loaded box, and the transmission line box. There are also many different configurations of subwoofer enclosures that are outside the scope of my project. In the papers that Dr. Richard Small presented to the Audio Engineering Society in 1972 and 1973, he suggested a very easy way to make an initial determination for what kind of enclosure to use with a driver. His suggestion was that by finding the efficiency bandwidth product, or EBP, you could determine what kind of box the driver would do best in. While not a steadfast rule, this suggestion has proved to be a good design guideline.

The equation for the EBP is:

$$EBP = \frac{F_S}{Q_{ES}}$$

Equation 2 - Efficiency Bandwidth Product

The guideline says that if the EBP is greater than fifty, then the driver will excel in a vented box. If the EBP is less than 50, the driver will excel in a sealed box. The corollary to this is that the driver should also have an X_{MAX} of at least 5mm for 10" to 12" cones, and at least 2mm for 6" to 8" cones. This is because in a closed box environment the woofer is required to make large excursions more frequently than in a vented box.[9]

The closed box is the simplest loudspeaker enclosure, consisting of an enclosed volume of air and a driver. The sealed enclosure acts as a second-order high-pass filter where the resonance and the associated damping, or Q, controls the response. Because of the easily controlled response and relative ease of construction, this box is especially good for beginning builders.[10]

The three parameters that determine the behavior of the closed box are f_s, V_{AS}, and Q. The parameter f_s denotes the driver's resonant frequency in free air. Just as any mechanical structure that exhibits periodic motion, there is a frequency at which the driver resonates most easily and stores the most energy. V_{AS}, as previously discussed, is the volume of air having a mechanical compliance equal to the compliance of the suspension system. Q is a slightly trickier concept to define. It

---

[9] Alden 15
[10] Dickason 29

is essentially a measurement of the control that the speaker possesses over its own resonance point. Generally speaking larger magnets have a smaller Q, provide a larger damping force, and have a cleaner transient response. A larger Q corresponds to smaller magnets and a smaller damping force. Several different values of Q are found throughout the driver, with Q_{TS} being the sum of the different values in free air. That sum is defined as below.

$$Q_{TS} = \frac{Q_{ES} * Q_{MS}}{Q_{ES} + Q_{MS}}$$

Equation 3 - Qts

To begin finding the dimensions of a sealed box is finding the value F_{CB}. When a driver is placed in a sealed box, it is easy to see how the compliance of the driver's suspension system would add with the compliance of a sealed air mass to result is what is essentially a stiffer suspension system. This raises the resonant frequency of the driver to a new frequency F_{CB}. This frequency can be calculated as below:

$$F_{CB} = \left[\sqrt{\frac{V_{as}}{V_b} + 1}\right] * F_S$$

Equation 4 – Fcb for a Sealed Enclosure

Likewise the Q_{TS} value will change when the driver is placed into the sealed box. The formula to calculate this change is very similar to the one for F_{CB}.

$$Q_{TS} = \left[\sqrt{\frac{V_{as}}{V_b} + 1}\right] * Q_{TS}$$

Equation 5 – New Qts When Placed in Sealed Enclosure

To then calculate the new rise at the resonant point compared to the reference level of the driver's response the following equation can be used.

$$Peak(dB) = 20log\sqrt{\frac{Q_{TC}{}^{4}}{Q_{TC}{}^{2} - 0.25}}$$

Equation 6 – Peak Amplitude of Sealed Enclosure

The corresponding frequency of the Peak dB can then be calculated.

$$F_{R\,MAX} = \left[\sqrt{\frac{1}{1 - \frac{1}{2(Q_{tc}{}^{2})}}}\right] * F_c$$

Equation 7 – Frequency of Peak Amplitude Point

To calculate the point at which the frequency response has fallen 3dB below the reference level the equation below is used.[11]

$$F_3 = \left[\sqrt{\frac{\left(\frac{1}{Q_{tc}{}^{2}} - 2\right) + \sqrt{\left(\frac{1}{Q_{tc}{}^{2}} - 2\right)^{2} + 4}}{2}}\right] * F_C$$

Equation 8 - -3dB Point of Sealed Enclosure

Using the equations given above, a fairly accurate picture of the performance of a woofer in a sealed box can be determined. More accurate simulation techniques will be discussed later.

The second type of enclosure that I will discuss is the vented or bass-reflex box. When I calculated the EBP as shown above for the woofer that I used in my

[11] Alden 24-28

design, the result was in the sixties. This meant that the woofer I used could be used in either design with a measure of success, but was slightly more appropriate for a vented enclosure. The vented enclosure also provided a lower frequency extension, which these woofers benefited from.

In a vented enclosure, the addition of a small port to the sealed air mass causes it to function as a Helmholtz resonator. When you blow across the top of a soda bottle and cause it to whistle it is the same acoustic event as a vented enclosure. When you blow into the neck of a bottle or jug, it compresses the air inside that smaller area. The air inside the bottle then presses back against the compression. The momentum of the air in the neck of the bottle then rarifies the air in the bottle, causing it to suck the air back in slightly. This back-and-forth motion of the air in the neck of the bottle has a natural resonant frequency just like the other resonant structures that I have discussed.

There are several ways in which this resonance can be used to an advantage. One advantage is that a smaller vented box can have an F_3 value equal to that of a much larger sealed box. Likewise, you could also design a vented box to have an F_3 value a third of an octave below a similarly sized sealed box. An additional benefit is that at the resonant frequency of the box the driver itself is actually moving very little, with most of the work being done by the Helmholtz resonator mechanism. This reduces the excursion of the woofer at low frequencies, and also the bass distortion.

There is also a set of disadvantages that goes along with the above reasons in favor of a vented box. It is more critically important with a vented box that the mathematics are correct. This includes any miscalculation or estimating on the part of the manufacturer. Ray Alden advocates that builders do their own measurements on the individual drivers to ensure proper performance. The boxes are also susceptible to leaks.[12]

Having chosen to pursue this design for my speakers, I followed the procedure for determining volume, F_3, F_B, and the length of the port for the flattest response I could generate with the chosen alignment. To start by finding the internal volume required, the equation below is used.

$$V_B = 15 * V_{AS} * \left[{Q_{TS}}^{2.87}\right]$$

Equation 9 – Target Volume for Vented Enclosure

The F_3 of the system is then calculated.

$$F_3 = \frac{0.26F_s}{(Q_{TS})^{1.4}}$$

Equation 10 - -3dB Point for Vented Enclosure

To find the frequency that the vented box should be tuned to the following equation is used.

$$F_B = \frac{0.42F_s}{(Q_{TS})^{0.9}}$$

Equation 11 – Vented Enclosure Tuning Frequency

Knowing those three parameters, the only remaining calculation is to determine the port length and diameter. Ray Alden suggests the following guidelines for vent diameter based on woofer size.

---

[12] Alden 32

Vent Diameter	Woofer Diameter
1"	4"V
2"	5"-6"
3"	6"-8"
4"	8"-10"
5"-6"	12"-15"

Table 1 – Suggested Minimum Vent Diameter Based on Woofer Size

The reason for these suggestions is primarily because of the amount of air moved by the woofer. As the size of the woofer increases, the amount of air moved through the vent also increases. When too much air is pushed through a small opening, it produces corollary effects to the Helmholtz resonator and a departure from theoretical performance.

The final calculation for port length can then be completed after having chosen an appropriate diameter from above. The equation for port length is as follows.[13]

$$L_V = \frac{1.463 * 10^7 * r^2}{{F_B}^2 * V_B} - 1.463r$$

Equation 12 – Vent Length

To apply those calculations to the woofer that I chose yields the following results shown in the order the equations were given above.

1.) $V_B = 15 * V_{AS} * \left[{Q_{TS}}^{2.87}\right] = 15 * 24.5\mathrm{L} * \ [.5^{2.87}] = 50.27\ L = 3067.61\ \mathrm{in}^3$

[13] Alden 32-35

2.) $F_3 = \frac{0.26F_S}{(Q_{TS})^{1.4}} = \frac{0.26*40.4}{(.5)^{1.4}} = \frac{10.504}{.379} = 27.72\text{Hz}$

3.) $F_B = \frac{0.42F_S}{(Q_{TS})^{0.9}} = \frac{0.42*40.4}{(.5)^{0.9}} = \frac{16.968}{.5339} = 31.66\text{Hz}$

4.) $L_V = \frac{1.463*10^7*r^2}{{F_B}^2*V_B} - 1.463r = \frac{1.463*10^7*.6875^2}{31.66^2*3067.6} - 1.463*.6875 = 1.24\text{in}$

It is worth pointing out that the vent I chose was slightly smaller than recommended. I was drawn to it because it was adjustable, but I didn't foresee how short I would need it to be.

Keeping in mind that the design of most studio speakers is to place the drivers on a face that is only slightly wider than the driver, I chose an initial width of eight inches for the front of my speakers. This is commonly done to give the drivers better imaging characteristics in the stereo field. Given the calculated internal volume I was trying to meet, I then made a few estimates and came up with the dimensions of eight-by-sixteen-by-twenty four. These internal measures would give me a volume of 3074, a mere .14% deviation from the number I was aiming at. I chose these internal measurements thinking that it would be easier to build something with round numbers, but this was actually a mistake. By allowing the dimensions to be direct integer multiples of each other, I also allowed an opportunity for standing wave development inside my enclosures. Two methods for avoiding this mistake are given by Alden. He names two ratios that are popularly used in planning dimensions. This first ratio is commonly known as the acoustic ratio and is (.7939:1:1.2599). The second is the commonly seen golden ratio

(.618:1:1.618). The method for implementing either of these ratios is to take the cube root of a desired volume and to multiply it by each number in the ratio.[14]

While the equations above give a good starting point and a rough estimate at the behavior of a driver in a vented enclosure, the most accurate prediction can be made with modeling software. For modeling the enclosure's effect on the driver's performance, I found Jeff Bagby's Frequency Response Modeler to be the most useful.[15] This program allows a .frd file to be imported for a driver and then manipulated with the response of the box that it is to be placed in. A .frd file is a data set of points containing frequency and SPL information giving a clear picture of the driver's response across the frequency range. While for human purposes it is easier to describe a driver's performance in terms of parameters and values that we can put into equations, a computer has no problem dealing with a discreet picture of a driver's performance comprised of measured data points. The Frequency Response Modeler allows the user to enter the parameters of their box to calculate the box response and the baffle edge diffraction as shown in figures 5 and 6.

---

[14] Alden 28

[15] http://audio.claub.net/software/jbabgy/jbagby.html

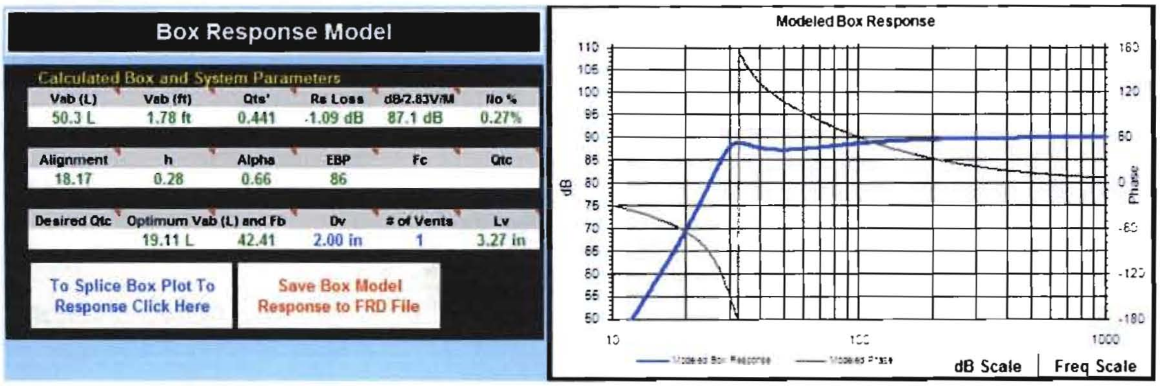


Figure 5 - Jeff Bagby's Frequency Response Modeler – Box Response Model Section

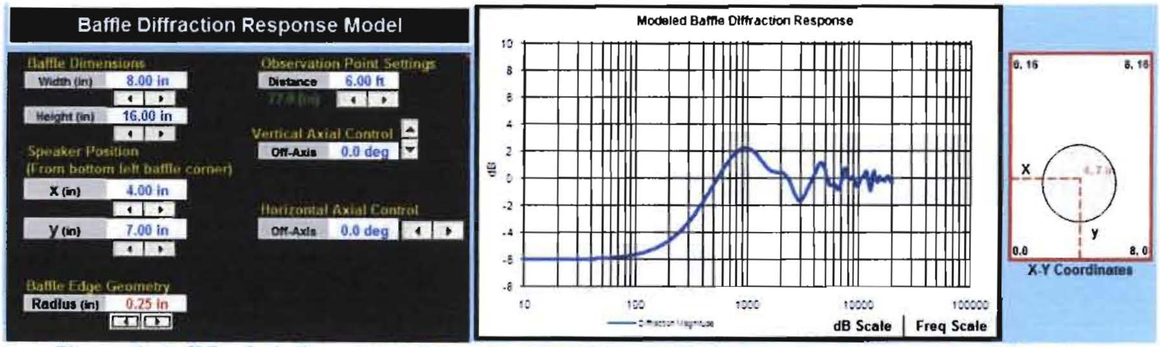


Figure 6 - Jeff Bagby's Frequency Response Modeler – Baffle Diffraction Response Model Section

The modeled response can then be superimposed on the .frd file or even spliced to the .frd file. Figure 7 shows the box response in grey on top of the driver response in blue.

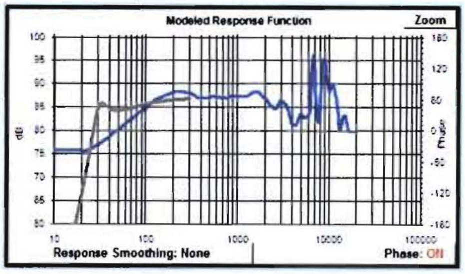


Figure 7- Response Plot with Box Response Superimposed

Figure 8 shows the box response added to the driver response up to a frequency of 300Hz. The program will then export the file representing the graph on the left so that when it is used in a crossover-modeling program, the program will have a more accurate picture of the driver and the enclosure.

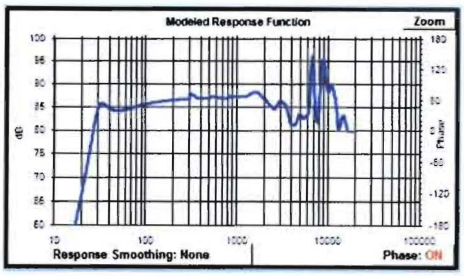


Figure 8 - Response Plot with Box Response Spliced

The Crossover

Having an accurate picture of what kind of enclosure I was going to use and the response of the driver allowed me to move on the final step of my design, the crossover. There are two main criteria used to describe the commonly encountered crossovers. Crossovers are usually described according to how many drivers they split the signal between, and the severity of the attenuation curve above the cut-off point. Most often in speaker building two-way or three-way crossovers are used. This means that the incoming signal is either split between a high and a low driver, or a high, a mid, and a low driver. More complex configurations are possible and are seen, but are rather uncommon due to the high amount of added cost and complexity compared to the small benefit. The severity of the attenuation above the cutoff point is described by the order of a filter. Common filters in speaker building go up to fourth-order, but rarely above, again for reasons of cost and complexity.

Passive crossovers always contain a combination of capacitors and inductors. These two electronic components exhibit frequency dependent resistance, also known as impedance. For a resistor, the resistance is the same no matter what the frequency. Capacitors function as a high-pass filter, offering high impedance to low frequencies and DC current. Inductors function as a low-pass filter, giving high impedance at high frequencies due to the back-EMF generated. The impedance value of each component is given as value *z* below.

$$z = \frac{1}{2\pi f C} \qquad z = 2\pi f L$$

Equation 13 – Impedance for Capacitor and Inductor

In the equations above, f is the frequency you wish to solve for, C is the capacitance value in farads, and L is the inductance value in henrys. Both the farad and the henry are very large units, so most values that you encounter will be microfarads or millihenries.

Another issue worth noting about impedance is that it is a complex value. An impedance value of a component, like an inductor or a capacitor, is actually a vector value with a phase component. This means that as you send your signal through the crossover the phase will be altered. When modeling a crossover it becomes very important to have an .frd file with phase data included because of this. A lack of phase data can produce results that aren't going to translate into the real world.

Knowing that a capacitor serves as a high pass filter, and that an inductor serves as a low pass filter, the circuit for a first-order two-way crossover is fairly simple to understand. In the diagram on the left we have just such a circuit. The top branch of the circuit essentially forms a voltage divider where the low frequencies encounter resistance at the capacitor and are dropped there, while the high frequencies are allowed to pass to the tweeter. The inductor likewise is where most of the high frequencies are

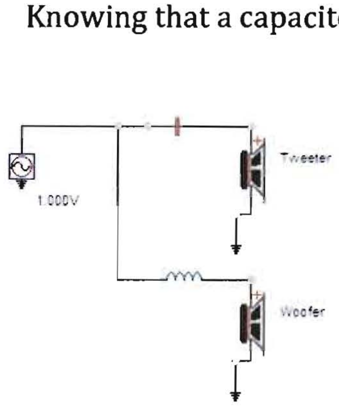


Figure 9 – First-Order Two-Way Crossover

dropped, allowing only the low frequencies to proceed to the woofer. A crossover like this would generate a 6dB/Octave drop past the cutoff point. The first-order crossover is the simplest and alters the phase the least, but is not usually practical due to the excessive amount of overlap required by the drivers. To calculate the values needed by the components in a first-order crossover, set the impedance of the capacitor or inductor equal to the impedance of the speaker at the desired crossover frequency and solve. The equations for that are shown below.

$$C = \frac{1}{2\pi f R_H} \quad L = \frac{R_L}{2\pi f}$$

Equation 14 - Capacitor and Inductor Values in First-Order Crossover

The second-order crossover is slightly more complex. In this case the addition of an inductor or capacitor in parallel with the driver acts as a shunt to ground and steepens the cut-off slope. One issue that appears in second-order crossovers that does not appear in first-order filters is that of the Q value. The Q value of a filter determines the type of the filter. With a first-order crossover, the only possible type is a Butterworth filter. For a second-order filter the equation to give the Q value is given below.

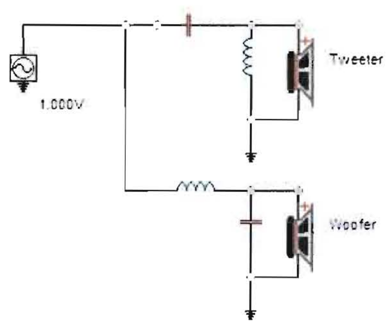


Figure 10 - Second-Order Two-Way Crossover

$$Q = [(R^2C)L]^{1/2}$$

Equation 15 – Q Value for Second-Order Filters

The following Q values produce the associated types of filters.

Q=.707	Butterworth
Q=.58	Bessel
Q=.49	Linkwitz-Riley
Q=1	Chebychev

Table 2 - Q Values and Associated Filter Types

The following equations then demonstrate the correct method for determining the values of the capacitor and inductor in the tweeter branch of a second-order filter, keeping in mind the Q value.

$$C_t = \frac{Q_t}{2\pi f R_t} \quad L_t = \frac{R_L}{Q_t 2\pi f}$$

Equation 16 - General Form for a Second-order crossover

The same approach applies to the woofer branch of the crossover, and from these general forms of the equation, an equation for any of the second-order filters can be easily derived.

The third and fourth-order crossovers are really no more than cascades of the previously mentioned circuits. The third-order crossover, as shown to the left, cascades another component in series with the driver. These filters will have an attenuation of 18dB/Octave after the cutoff frequency. A third-order filter is by definition

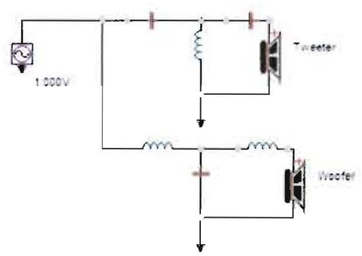


Figure 11 - Third-Order Two-Way Crossover

a Butterworth filter, and in fact this is the only option for all odd-order filters.

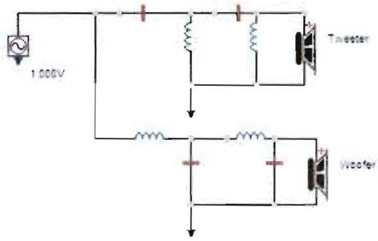


Figure 12 – Fourth-Order Two-Way Crossover

The fourth-order filter is a complete duplication of the second-order filter, and yields a loss of 24dB/Octave after the cutoff. Because of the increased complexity of this circuit it has even more filter types than the second-order. Some of these include the Legendre, Gaussian, and Linear-Phase filter.

The three-way crossover contains many of the same elements as the two-way, but multiplies in complexity with the addition of extra variables. The figure to the left shows the basic idea of a first-order three-way crossover. The high-pass and low-pass filters can be combined to create a band-pass filter, which is then applied to the midrange. As you can see from the figure, this immediately doubles the parts count in the crossover circuit. As I didn't use this design in my project, I will shy away from a detailed discussion of the three-way crossover.

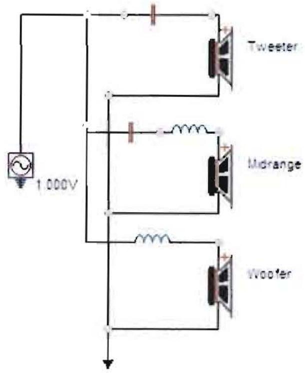


Figure 12 - First-Order Three-Way Crossover

When designing my crossover, I started with the equations for a second-order Linkwitz-Riley crossover. After I had these initial values I entered them into a crossover simulation program called lspCAD. There are many software programs, both free and paid, that will do this job. I unfortunately cannot recommend any programs besides lspCAD, as it was the only program that I used.

The textbook formulas are an excellent starting point for designing a crossover, but I honestly found a fair amount of guess-and-test was required to achieve the response that I wanted after I had built my circuit in lspCAD. After several rounds of posting my design on the Parts Express forum and receiving criticism and feedback, I arrived at the design to the left. The final design ended up having a third-order filter on the tweeter with an additional filter known as a zobel network to roll off the high end. The low branch of the crossover has a second-order filter with a small tank capacitor in parallel with the inductor. The tank capacitor serves to take out the higher breakup modes discussed earlier. The projected frequency response is shown to the right.

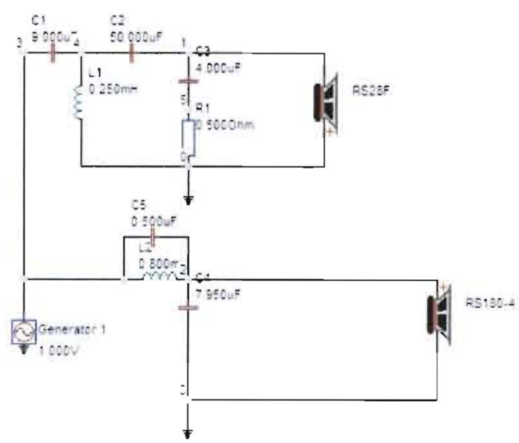


Figure 13 - Final Crossover Design for My Project

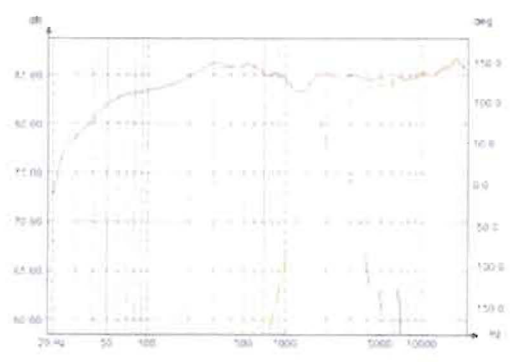


Figure 14- Projected SPL Response

Construction

Following the plans discussed in the above sections, I started actually constructing my project. My woodworking skills are fairly novice, and the woodwork is a very significant part of building speakers. It would be very fair to say that I severely underestimated this part of the process.

As per the recommendation of countless sources, I planned to use medium-density-fiberboard as the medium for my project. This pressed-wood product is very dense and helps dampen any vibrations from taking hold in the wood itself. The bulk of this wood however makes it hard to work with, and it is very dusty. I would highly recommend working with MDF in a well-ventilated area and wearing a dust mask.

I cut my 4'x8' panel to the sizes of my speaker walls using a circular saw, as

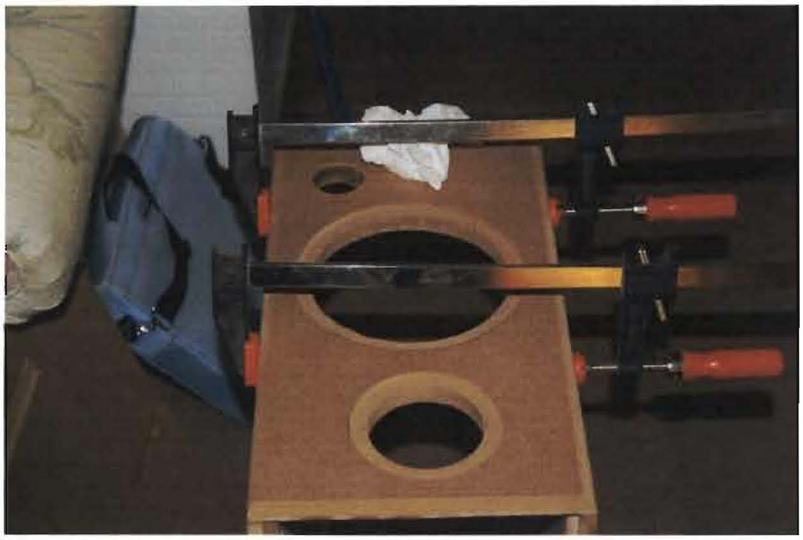
shown to the left. A circular saw turned out to be a poor choice of tool for this task, and to create a properly fitted cabinet I should have used a table saw. Eventually I would find that my sides were not very square when I went to assemble the cabinets.

I then took my pieces to the College of Architecture and Planning wood shop

to complete the rest of my woodwork. I added a 3/8" half-lap to the sides of all of the pieces with a table saw so the wood glue would bond more securely. The tweeter hole was accomplished with a fly cutter, and counter-sunk with a ½" rabbeting bit. The woofer hole was made with a jigsaw and a ½" rabbeting bit. The hole for the vent was not counter sunk, so it was made with a hole saw. Likewise the hole for the terminal cup on the back was made with a hole saw. The edges of the front and back were rounded with a 3/8" half-round bit on a routing table.

To begin assembling the box I first secured the terminal cup to the back of the box with hot glue.

I then started gluing the pieces of the box together one at a time, as I only had two clamps.



While waiting for the glue to cure, I then started on the crossover. I used a perforated project board from Radio Shack to mount all my components, securing them with hot glue.

After the components were secured and soldered together properly I

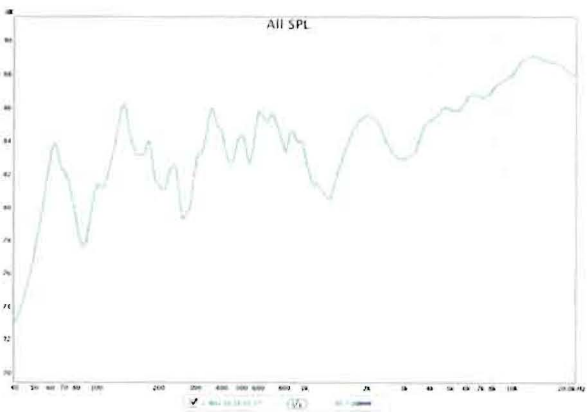
proceeded to test the crossovers to make sure that they functioned properly before I sealed them in the enclosures.

When I found that both crossovers were working properly, I secured them to the bottom of the box which had not been attached yet. I installed the drivers into the front baffle and soldered their leads to the crossover. I then soldered the leads from the terminal cup to the crossover as well. With all of the wiring complete I glued and clamped the final piece of each box into place. Since my cuts weren't perfect I then went back and sealed any gaps in the corners of my box with a plain silicone caulk. After letting the caulk set for about three days, my speakers were ready to be heard.

Results

After my project was complete, I proceeded to do a series of listening tests with material that I knew well. My objective evaluation is that they perform very well on bright subject material. The tweeter is very bright and incredibly present, but not to the point of unpleasantness. Source material such as acoustic guitar, jazz piano, and voices seem to really shine. The bass is impressive to me since I know the response of the woofer without the vented enclosure, but they don't excel at bass heavy music. The balance across the frequency spectrum seems to be good to my ears, and the imaging is good but not amazing across the soundstage. The biggest regret that I have is that these speakers are huge. The depth of 24" is excessive, and makes them hard to put anywhere. If I had thought more about the design I might have arrived at a floor-standing version that was 8" deep and thrice as tall.

After running a frequency response test on one of the cabinets I got the



frequency response curve shown here Apart from the dips at around 85Hz and 250Hz the graph looks much like I had calculated. Due to the non-anechoic nature of the room that I was testing in there were going to be reflections off of the wall which are what is seen in those two dips at 85Hz and 250Hz. Overall I was very happy with the way that the results came out and the way that the speakers sounded.

Bibliography

Alden, Ray. *Speaker Building 201: with 11 Completely Designed Speaker Systems including a 5.1 Home Theater System*. Peterborough, NH: Audio Amateur, 2004. Print.

Dickason, Vance. *The Loudspeaker Design Cookbook*. Peterborough, NH: Audio Amateur, 2006. Print.

Reflection

Looking back at this project I am very happy with my choice of topic. I feel that I learned so much during this project, and it really connected my knowledge with a tangible task that I was equipped for. I do think looking back that this was an exceptionally large project that I couldn't quite see the scope of at the beginning, but now that it's done I have no regets about tackling something this large.

My initial approach to the project started last spring when I was brainstorming what I might want to do. So by the time that this semester came around I had already commited several months of casual research to the topic. I thought that I had the theory fairly well understood, but after building the speakers I came to realize that a little hands-on experience can quickly show you many things that you'll never get reading research.

As far as the fabrication process went I did find that very frustrating. I had so little woodworking experience that I did not understand how hard it was to put together a square box. In the end I made it work, but they are not attractive. My perfectionist nature reared up quite a few times while building them and almost caused me to start over. I persevered though in the hope of finishing my senior project and came out with a product that I am moderately proud of.

There was one thing however that I hoped to incorporate into this project that had to be omitted for the sake of time. I had started designing a website that was to be a comprehensive walkthrough of the building process. I spent quite a lot of time learning css and html, but unfortunately the semester ran too short for me to accomplish that. I still plan on continuing with that website after the semester is over though to continue learning those web coding skills.

Overall I am very happy with my senior capstone, and it was a good exploration into something that I want to continue to do. The help that I received from my advisor was invaluable. I hope to be able to continue refining my woodworking skills and to be able to create projects in the future that can build and expand on the work that I've done here.